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VoIP in an Internet service provider network

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IN THIS ARTICLE, I TAKE A LOOK AT how "traditional" Internet Service Providers deploy Voice over IP (VoIP) [1] in their networks. VoIP gets a lot of press these days, and there seems to be no end to the hype surrounding it, especially when it comes to service provider networks. This article makes no attempt to cover the basics of VoIP (see Heison Chak's excellent column here in *;login:*), as this is a huge subject and there are plenty of Web sites available for learning more.

There are at least two ways "traditional" Internet service providers could use VoIP in their networks. The first (and arguably the oldest) method is as a replacement for voice trunks (trunking, either point-to-point or long distance) in their voice networks. This might be done to reduce cost or enable or enhance new voice services. The second way VoIP is typically deployed is as a replacement for a traditional landline phone (for example, a Vonage [2] or Packet8 [3]) type of service).

VoIP Trunking

Trunks are telephone lines that are used for point-to-point or long distance calling on provider networks. They carry calls from one point-of-presence (POP) to another, or from the POP nearest to the recipient to the recipient.

The use of VoIP connections as trunks has actually been around for a relatively long time. Many traditional long distance providers started using VoIP on their *dedicated* IP networks 10 years ago (or more) in a bid to lower cost. The key word here is "dedicated," because without a quality-of-service component, running VoIP on highly utilized non-QoS–enabled IP networks is usually a recipe for trouble. Once a QoS mechanism was designed and implemented, it made running VoIP on non-dedicated networks a more feasible proposition.

Providers might route calls onto their VoIP network in order to get them closer to the recipient's end office, thereby reducing cost. Once the call is closer to the recipient, the call would then be converted to a traditional voice trunk and handed off to the local carrier who has the callee (recipient) as a customer.

More recently, providers can route calls directly to VoIP trunks by purchasing VoIP trunks from a provider such as Junction Networks [4] or Covad [5]. Alternatively, with the proper interconnection (peering) agreements and facilities, providers can route calls directly to their caller's destination network with little (or no) cost. Of course, the interconnection (peering) agreements in question may or may not cover VoIP. These peering agreements are usually made in cases of relatively equal traffic. A provider with a small amount of traffic probably cannot make a peering agreement (for VoIP or anything else, for that matter) with a much larger network unless something else changes (for example, money changes hands).

End-User VoIP Service

A provider might want to provide traditional Plain Old Telephone Service (POTS) access for all inhabitants of a large, multi-unit building. One way to do this would be to deploy a VoIP PBX such as Cisco CallManager (CCM) [6] to support local phones in the building. The CCM would terminate VoIP calls for subscribers and send the calls (via traditional closet RJ11/RJ45 wiring) to the recipient's subscriber phone. This allows the provider to aggregate and oversubscribe traditional voice trunks, either locally or across their WAN. This model is closest to the traditional POTS most subscribers are used to.

One important point to bring out regarding CCM is the fact that it uses Cisco's proprietary Skinny Client Control Protocol (SCCP) [7] to communicate between CCM and the VoIP phone. This might cause interoperability issues if the provider were to change PBX platforms, although other vendors do support this protocol.

In the Packet8/Vonage type of service, the end subscriber uses what is known as an Analog Telephone Adapter (ATA), which converts a traditional analog phone to VoIP. This type of service gives the end subscriber the most services and flexibility, as it pushes the VoIP functionality and benefits as close as they can get to the subscriber. These services use the Session Initiation Protocol (SIP) [8].

The benefits to a Vonage-type service are that additional services can be offered to the subscriber and the service is portable. To move service from one location to another, the ATA is simply moved. The downside to this method is that the adapter requires 110 volt power and 911 service can be difficult, if not impossible, to offer with current technology in widespread use.

Provider-Side Equipment

Of course, there must be equipment on the provider side to process and route calls initiated by subscribers and provide other services such as voicemail. If a provider has an existing "legacy" telephone switch (such as a Lucent 5ESS), it can support VoIP call termination with an appropriate card installed. Although this works, typical traditional wireline switches don't have the functionality in terms of end-subscriber features found in more modern softswitches [9] such as Asterix [10] or Metaswitch [11].

Asterisk is an open source softswitch sponsored by Digium that runs on Linux and handles POTS and trunk (T1) lines with appropriate hardware [12]. Asterisk is widely used because of its large feature set and its low cost, running on Linux and similar open source platforms. The one downside to Asterisk is the difficulty in configuring the software for use, as it must be done by text file and the process can be quite time-consuming. There are a number of GUI interfaces for Asterisk; voip-info.org has a very comprehensive list [13] of Asterisk front ends.

Metaswitch is a carrier-based class 4/5 switch replacement typically deployed in a provider environment. Softswitches are very attractive to providers owing to their lower space, power, and cost requirements (not to mention subscriber feature sets) compared to legacy telephone switches. In provider networks that already have a class 4/5 switch, they are deployed alongside the existing switch. This is usually done for financial reasons, as the class 4/5 switch has paid for itself many times over and is cheap to run.

Billing

No article on deploying VoIP in a service provider environment would be complete without a mention about how to collect money from the subscriber. The requirements of the billing system are directly dependent upon the plans offered by the provider. Of course, if the provider is only offering a flat-rate service with no international calling component, then billing is much easier, as no call detail records (CDR) need to be processed.

If a service provider has a billing system that handles existing voice services, then it is not a huge issue to add a time-based VoIP service offering. All major softswitches provide the standard CDR that is easily processed by a voice-based billing system such as Oracle Infranet [14]. See voip-info.org for a nice listing of VoIP billing systems [15].

Summary

VoIP is a great way for traditional Internet service providers to lower costs and offer new services to their subscribers. VoIP trunks are a good way to reduce more traditional T1 long distance and point-to-point trunk costs while maintaining an acceptable QoS level, either across their own net or on shared networks. Providing end users with telephone service can be accomplished by using a VoIP PBX for multiple dwelling units or via the ATA adapter. Equipmentwise, providers can use open source solutions or commercial class 4/5 softswitches, depending on their level of comfort with open source and VoIP feature requirements. On the billing side, VoIP integration is dependent upon what plans the provider opts for and what billing system the provider currently has in place. Most softswitches have standard CDR support, making billing integration relatively easy.

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